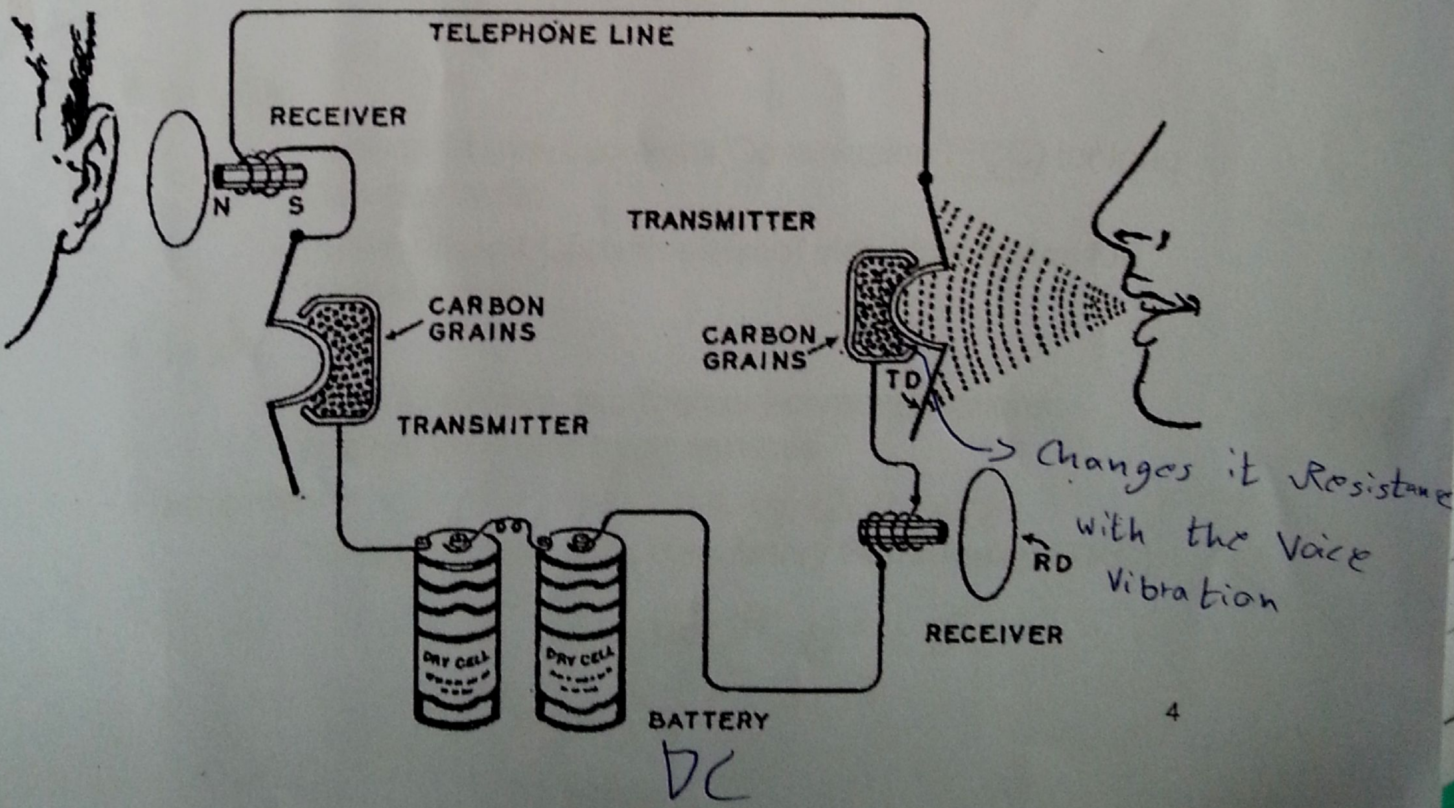
Com Systems Notes

DR . RA3ED ALZO3BIE
BY : YAMMAN KHTOUM

Analog Telephone Network

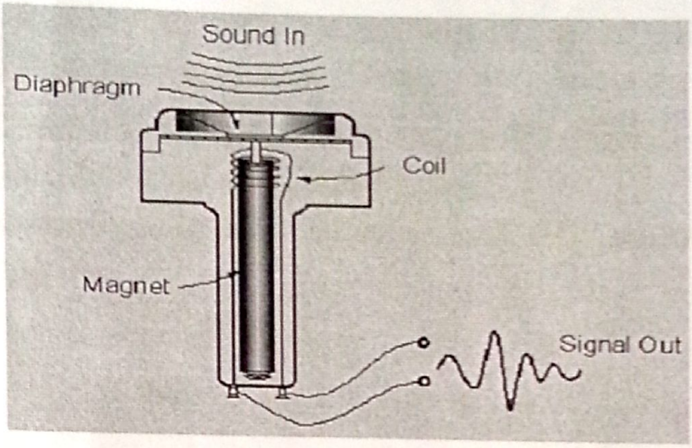
Telephone Circuit



The First Telephone

- In 1876, Alexander Graham Bell made the first Telephone, called the Bell Telephone
- The same equipment is used at TX and RX

Without Carbon Grains



1876 → Alex Gram Bell

Telecommunications Regulations

- In USA
  - » Federal Communications Commission (FCC) for long distance traffic
  - » Public utilities Commissions of individual states for local services
- In UK
  - » British Telecom and Mercury communications provide local and trunk services

↳ long Distance

- In Jordan
  - » Telecommunications Regulatory Commission (TRC)

هيئة تنظيم الاتصالات

Telecommunications Standard Organizations

- **In North America:**
- Bell Telephone Laboratories and Western Electric.
- The US Telephone Association (USTA).
- Exchange carriers standards Association (ECSA).
- Electronic Industries Association (EIA).
- Institute of Electrical and Electronic Engineers (IEEE). → makes too many standards
- Bell communications Research (Bellcore).
- The Federal communications commission (FCC): controls the radio spectrum in USA.

TABLE 1.2 IEEE Local Area Network/Metropolitan Area Network (LAN/MAN) Data Communications Standards → important for the manufacturer and the researcher

IEEE 802.1	Overview and Architecture, Bridging, Virtual bridged LAN (VLAN)
802.2	Logical Link Control (LLC)
802.3	Carrier Sense Multiple Access (CSMA) with Collision Detection (CD) (Ethernet)
802.4	Token Bus (Arcnet)
802.5	Token Ring (IBM Ring)
802.6	Queued Packet Synchronous Exchange (QPSX)
802.7	Broadband
802.8	Optical Fiber Technologies
802.9	Integrated Services
802.10	Security
802.11	Wireless
802.12	Demand Priority
802.14	Cable TV

can be found on the internet

America → IEEE
world → ITU

Telecommunications Standard Organizations

- **Outside North America:**
- The International Telecommunication union (ITU):

1. International Telegraph and Telephone Consultative committee (CCITT, now ITU-T): established recommendations for telephone, telegraph, and data transmission circuits and Equipments.

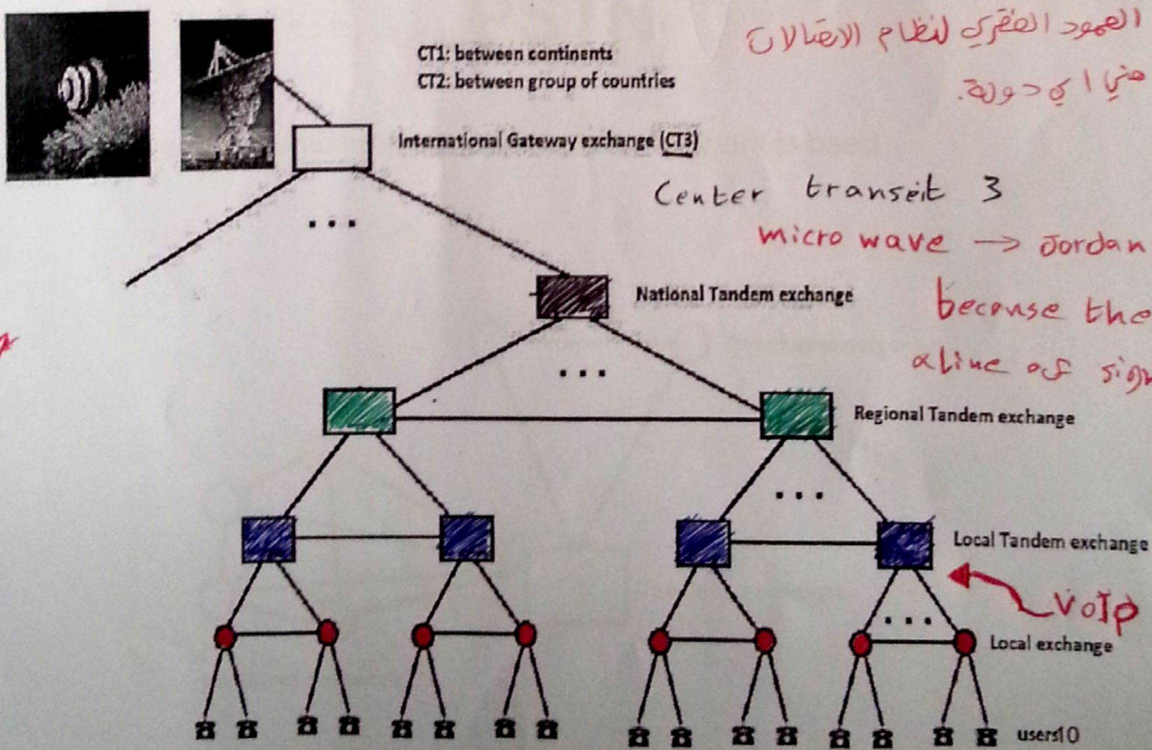
→ French word   FAX   → wire

2. International Radio consultative committee (CCIR, now ITU-R): coordinating the use of the radio spectrum.

Radio waves   → wireless

Public Switched Telecommunication (Telephone) network- PSTN

العمود الفقري لنظام الاتصالات
مثيا كدولة.



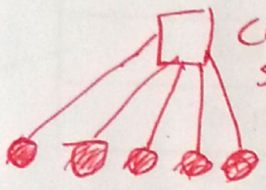
Center transeit 3
micro wave → Jordan and syria
because there is a line of sight

VoIP

PSTN

every node is connected directly to other node

- CT: centre de transit
- Connecting all the exchanges as a mesh network is called **junction network**
- Connecting all the exchanges as a star network is done via a central switching center called a **tandem exchange**
- Connecting Tandem exchanges is called **trunk network** (it is used between countries)



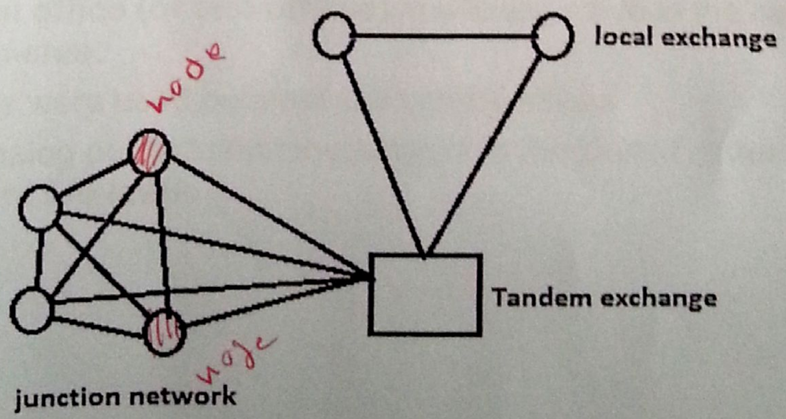
it's the main center that connects All the Exchanges

mesh → Advantage: * more liable, * more secure → dis Advantage: more cost (many connection)

star → not Rialble → disAdvantage
→ less cost → Advantage

PSTN

- In practice, a mixture of mesh and star network is used



Terminology

not for memorize

USA	UK (Jordan)
Customer loop	Local network → اول مقسم
Central office	Exchange
End office	Local exchange
Inter-office trunk	Junction
Junctor	Trunk
Toll office	Trunk exchange
Toll network	Trunk network

toll quality → enough Quality to be connected over long Distance

The Analog Network Hierarchy

- **Bell System Hierarchy:**

- Alexander Graham Bell invented the first practical Telephone in1876.
- **Central office (or end offices):** switching office in the center of a service area.
- **Trunks** were used between the central offices.
- The analog public telephone network in the United states evolved to a total of five levels.

Advantages
1) less cost

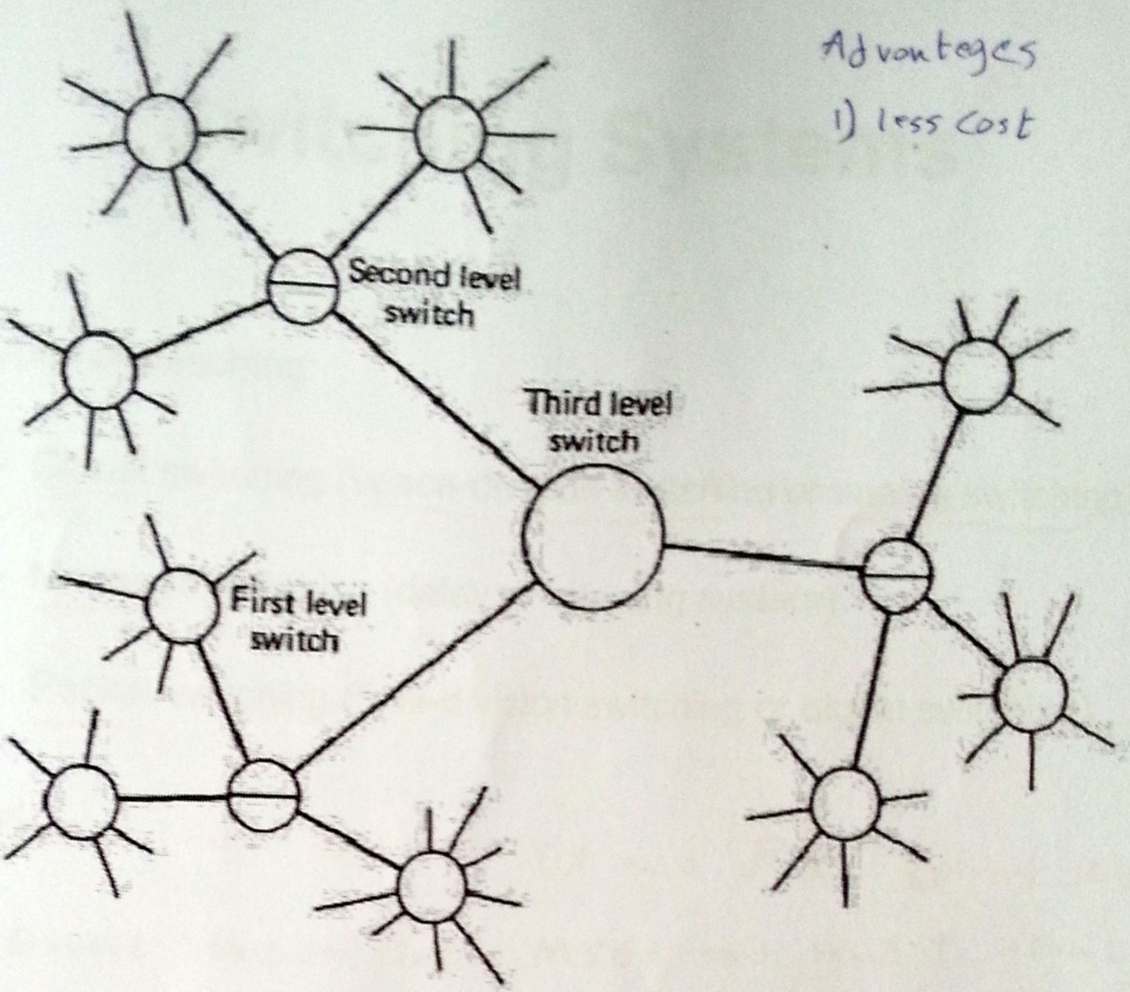
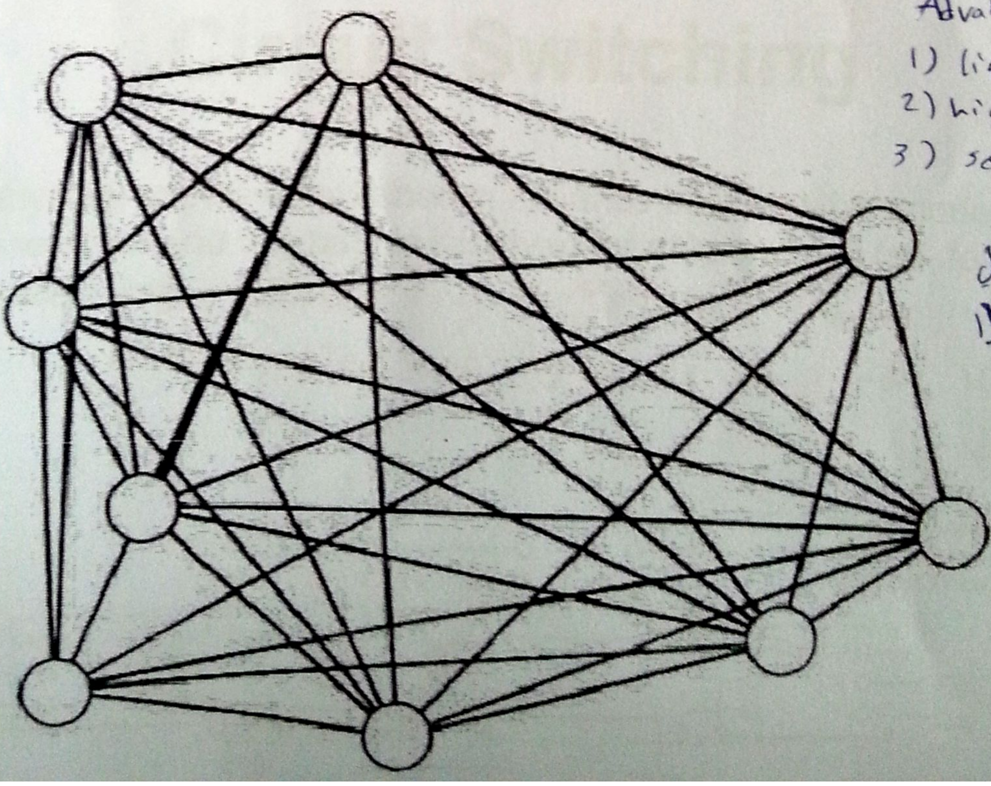


Figure 1.2 Three-level switching hierarchy.



Advantages
1) liability
2) high BW
3) security

disAdvantages
1) cost

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Switching Systems

- Types of switching:
  - Circuit switching (space-division switching or analog switching)
  - Message switching (delay or queuing system)
  - Packet switching (time-division switching or digital switching)

physical path between TX and RX (direct path)

convert the call into MSGs and send it without a direct path → we need a BW Available. (long MSG is Divided into shorter MSGs

Divide the MSG into a fixed packets of Data and send it through the office → TDM

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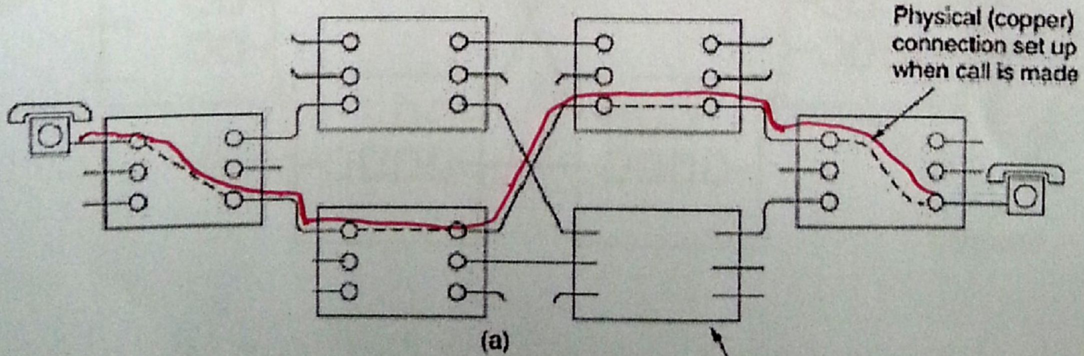
Circuit Switching

- A physical path is established in advance between the sender and the receiver and this path is reserved for only one call (so, for voice network) × BW Realibility is Bad → (hese moments)
  × Utilization of BW is Bad → Use Packet Switching



(a)

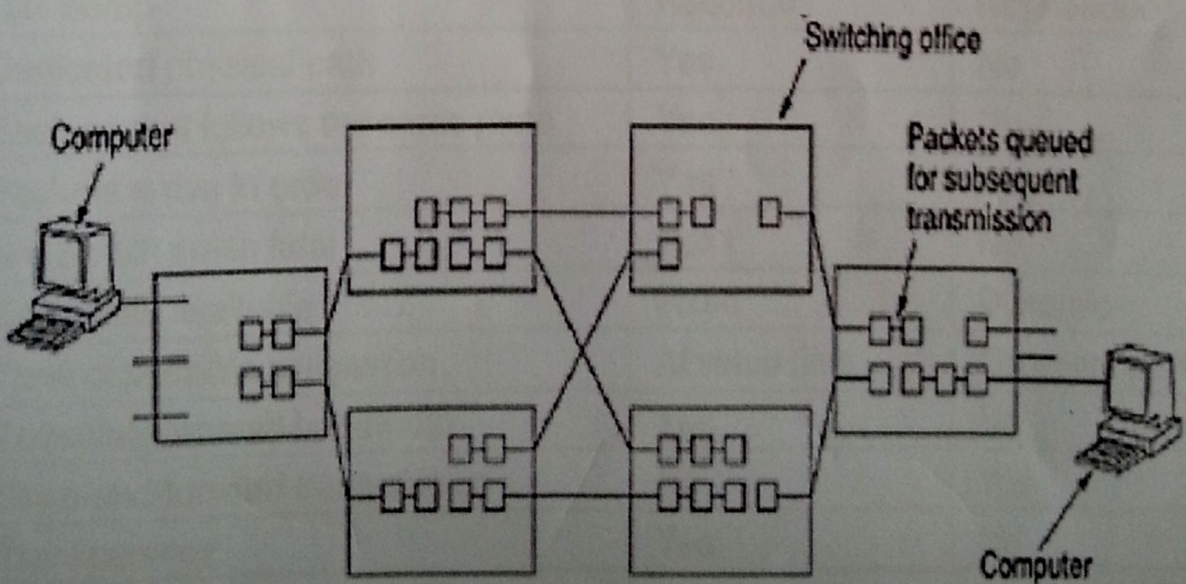
Packet Switching

- Dividing the long messages into packets is useful to solve the problem of queuing messages with different lengths
- Used in internet

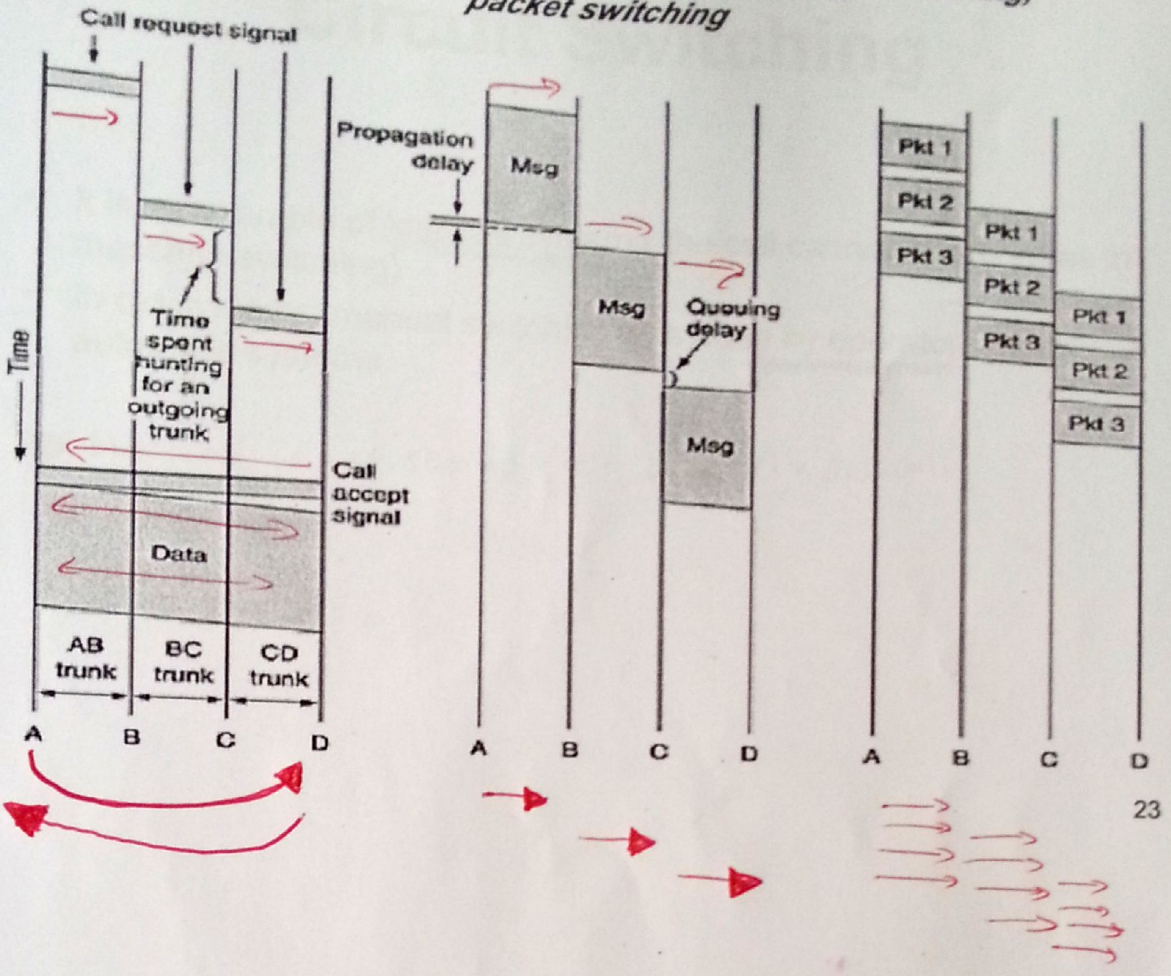
- Developed in 1969 by ARPANET (Advanced Research Projects Agency). So, ARPANET was the pioneer to today's Internet

- The Department of Defense wanted a more reliable network with route diversity capability. In a national emergency such as the September 11, 2001 attacks in the United States on the Pentagon in Washington, DC, and the World Trade Center in New York City, the Internet still functioned when many portions of the public voice and cellular networks were either out of service or so overwhelmed with traffic that people could not make calls → because the internet doesn't need a direct physical path → it just need a Routing ways to send / Recieve the packets

Packet Switching



...ming of events in circuit switching, message switching, packet switching



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* in packet switching → the Application layer is Responsible of Rearranging the Recieved ~~signal~~ packets and deliver it to the User

Item	Circuit switched	Packet switched
Call setup	Required	Not needed*
Dedicated physical path	Yes	No
Each packet follows the same route	Yes	No
Packets arrive in order	Yes	No
Is a switch crash fatal	Yes	No
Bandwidth available	Fixed	Dynamic
Time of possible congestion	At setup time	On every packet
Potentially wasted bandwidth	Yes	No
Store-and-forward transmission	No	Yes
Transparency	Yes	No
Charging	Per minute	Per packet

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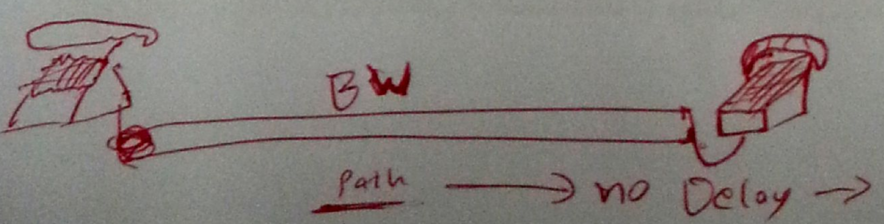

TV transmission fuck packets

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Circuit Switching

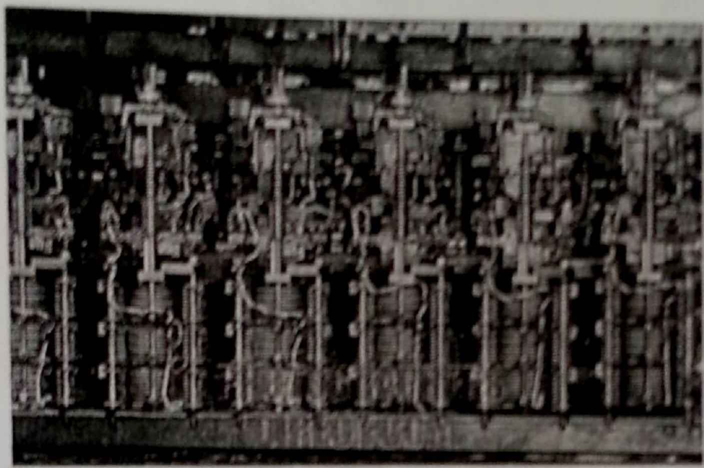
- It is an example of lost-call system (the call cannot be stored as in message switching)
- In old systems: manual switching was done by operator. Then, automatic systems.

the call is not stored it's Live Audio call

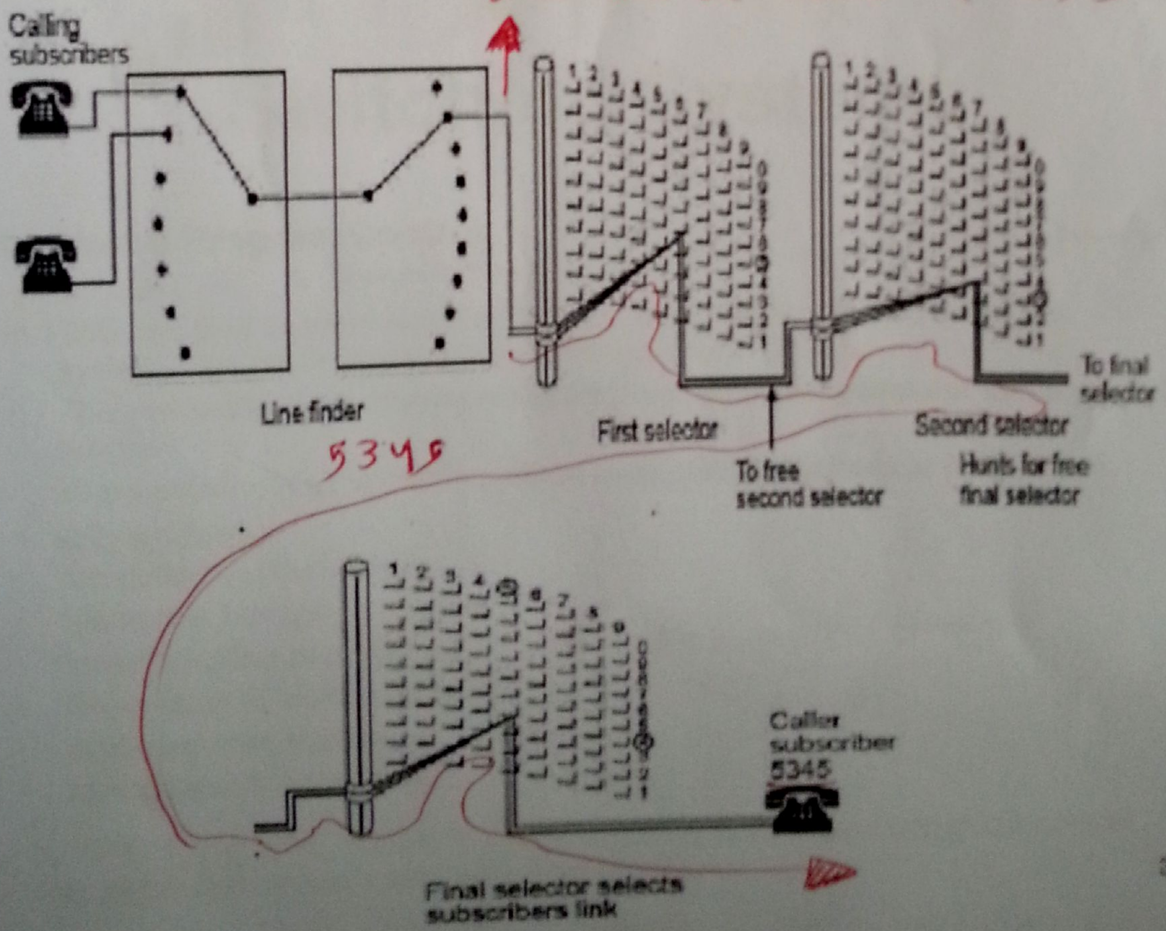
Switching Systems

- **Step-by-step switching system:**
- Uses the two-motion selector which was invented by Almon B. Strowger
- Had a lifetime of nearly 100 years
- It is the first automated switching system

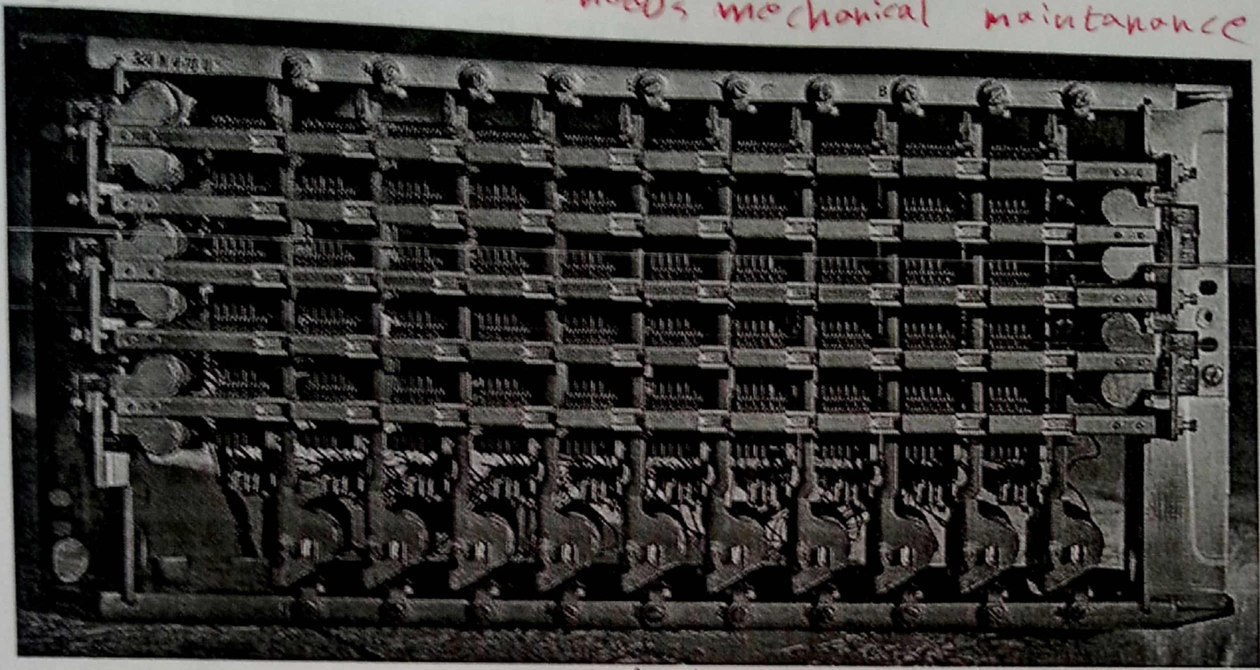
After operators

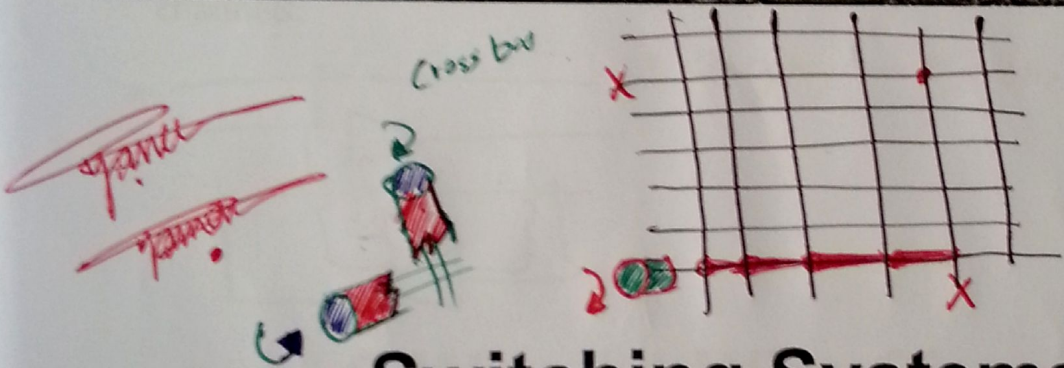


tune will be heard (صفارة)



Crossbar Switching System → needs mechanical maintanance





ما الهوش داخله ابدن

Switching Systems

b) Stored Program Control:

→ control is computerized But the matrix Cross point was still mechanical

In 1965: the Bell System installed the first computer-controlled switching system: The No.1 Electronic Switching System (ESS)

- This switching system uses a stored-program digital computer for its control function.
- The switching matrix of this system uses electromechanical reed relays
- **<u>Advantages:</u>**
- Simplified maintenance.
- Made number changes easy.
- lower blocking probabilities. → "Network Busy"
- Generation of traffic statistics.
- Automatic call tracing.
- Message unit accounting.

In 1976: A T&T's No. 4 ESS is a high-capacity switch used computer control and digital electronics for its switching matrix.

All computer Based controlled

Four-wire System

- The term four-wire has evolved to imply separate channels for each direction of transmission, even when wires may not be involved.

- For example, fiber optic and radio systems that use separate channels for each direction are also referred to as four-wire systems.

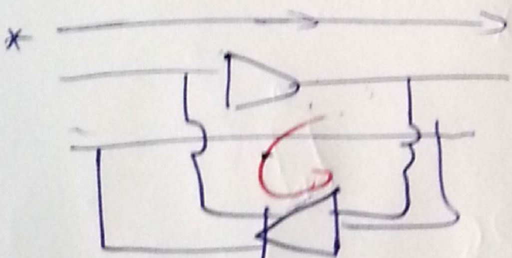
- Long distance transmission requires amplification and most often involves multiplexing.

- These operations are implemented most easily if the two directions of transmission are isolated from each other.

- Thus interoffice trunks typically use two pairs of wires or two fibers and are referred to as four-wire systems.

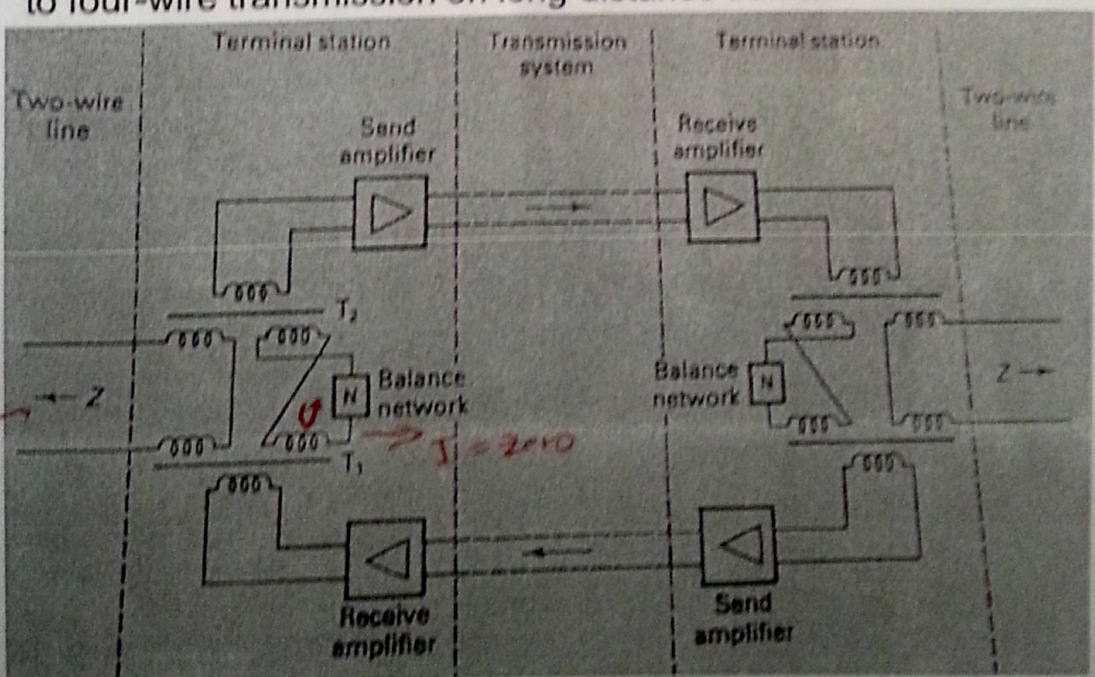
- a net savings in copper could result from multiplexing more channels.

Singing → echo over Long Distance



2-Wire / 4-Wire Circuit

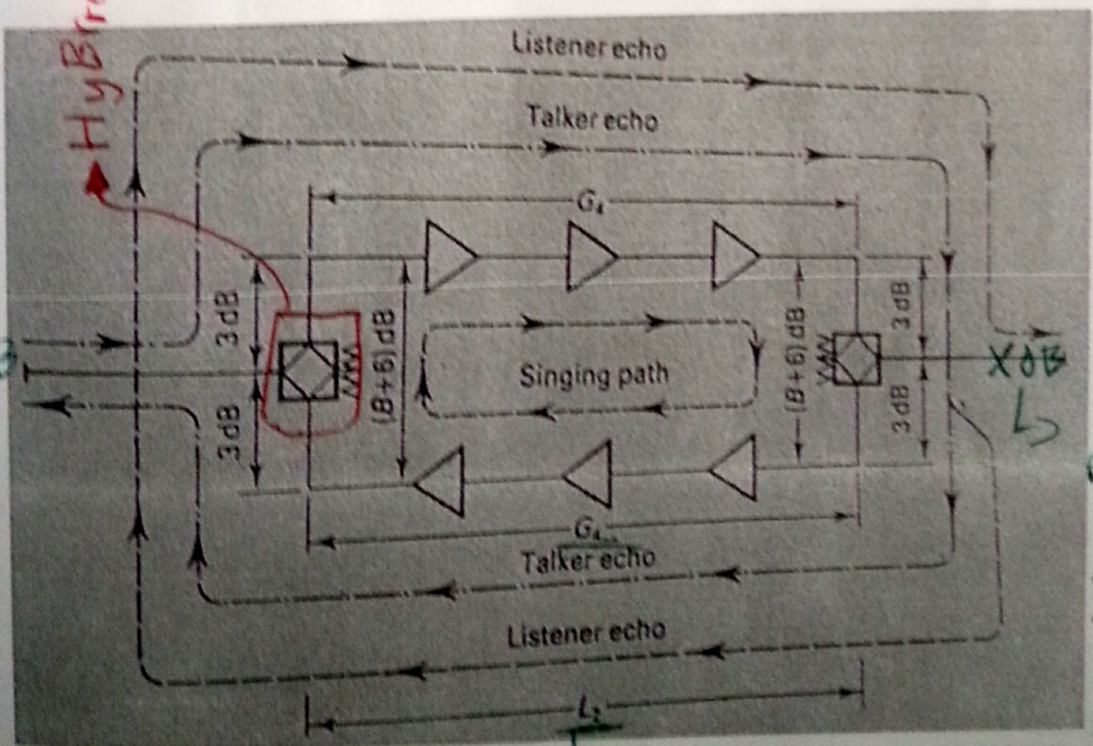
- It is necessary to convert from two-wire transmission of local loops to four-wire transmission on long-distance trunks.



to Get Balance → N ≈ Z

2-wire / 4-wire circuit

Hybrid transformer



Power at Y

$Y = X - 6 + G_4$

$= X + G_4 - 6$

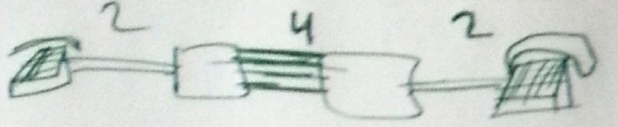
↳ $6 - G_4$

→ $6 - G_4$

3dB

3dB

(G_4)



2-wire / 4-wire circuit

incase of mismatch Between N and Z

- Total attenuation from one two-wire circuit to the other is

$$L_2 = 6 - G_4$$

- Transhybrid loss (TL): is the attenuation through the hybrid transformer from one side of the 4-wire circuit to the other.

TL = 6+B dB

$$B = 20\log\left|\frac{N+Z}{N-Z}\right| \text{ dB} \rightarrow N = Z \rightarrow \infty \text{ loss} \rightarrow \text{no echo}$$

22.5 ms is the Max accepted Delay

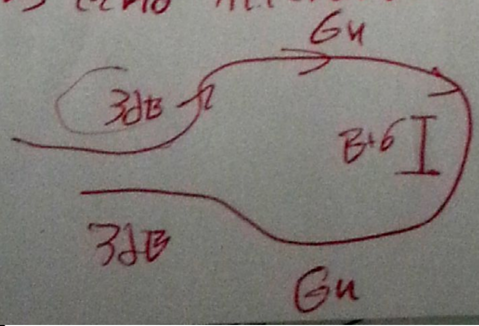
2-wire / 4-wire circuit

– B: balance return loss due to impedance mismatch between 2-wire line and the balance network
– Z: impedance of the 2– wire line
– N: impedance of the balance network

- The attenuation of the echo that reaches the talker's 2-wire line

$$L_t = 3 - G_4 + (B+6) - G_4 + 3 = 2L_2 + B \text{ dB}$$

L_t → echo Attenuation



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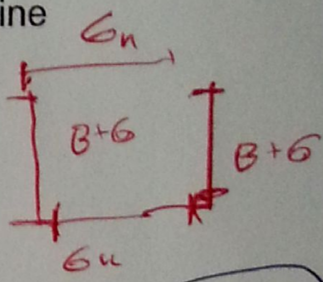
2-wire / 4-wire circuit

صوت مرتين

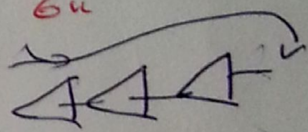
- The attenuation of the echo that reaches the listener's 2-wire line

بدنا كبيرة should be huge

$$L_1 = (B+6) - G_4 + (B+6) - G_4 = 2\underline{L_2} + 2B \quad dB$$



- We can control echo by applying loss when 2T**4** < 45 msec (increasing L2 by increasing the length but this increases delay)



Attenuating by length in the echo / Listener path

want L1 L2 LT to be big ⟶ high loss of echo

⟶ $L1 = 2L_2 + 2B$ ⟶ increase the line lenght.

* cant ADD Resistance because that will change the Coaxial cable characteristics

* Limitation ⟶ if the delay is less than 22.5 ⟶ then 22ms it's ok, your Ears won't tell the differanc

2-wire / 4-wire circuit

- Connections that produce more than 45 msec of roundtrip delay (representing 1800 miles of wire) require more attenuation for echo control than can be tolerated in the forward path. In these cases one of two types of devices was used to control the echo:

1. Echo suppressor
2. Echo canceller.

* Amplifiang the signal

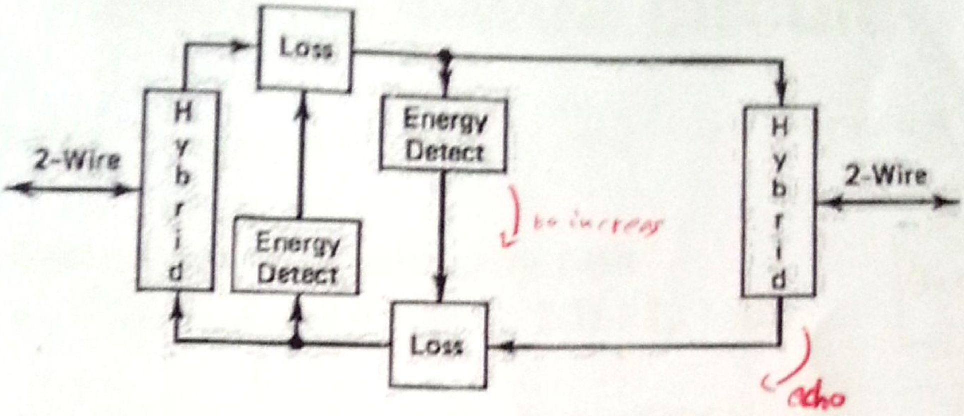


Figure 1.29 Echo suppressor.

* if the two (talker/listener) talk at the same time we'll face a problem

- One drawback of echo suppressor was that they might clip beginning portions of speech segments. If a party at one end of a connection begins talking at the tail end of the other party's speech.

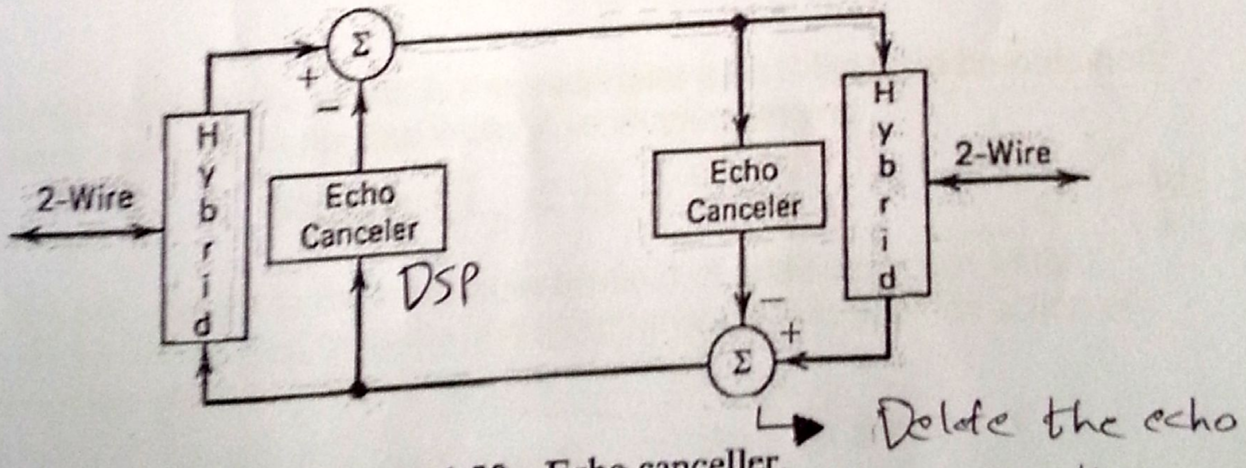
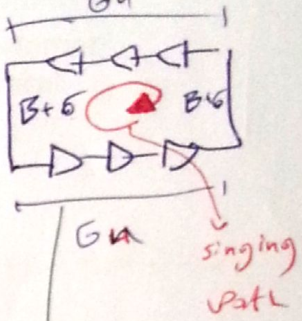


Figure 1.30 Echo canceller.

- It is better than echo suppressor.
- Because the cost of digital signal processing (DSP) technology has dropped so dramatically, echo cancellers are now used in any situation requiring echo control.

2-wire / 4-wire circuit

- **Stability:** echo cancalation and to prevent it from circulation



The net loss of the singing path

$$L_s = 2(B+6-G_4) \quad dB = 2(B+L_2) \ dB$$

singing

The necessary condition for stability is

$$L_S > 0$$

$$L_2 + B > 0$$

$$-L_2 < B$$

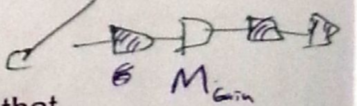
$$G_2 < B$$ → then its stable

2-wire / 4-wire circuit

- Singing point of a circuit is the maximum gain S that can be obtained from 2-wire to 2-wire line without producing singing

From $G_2 < B, \ \boxed{S = B}$

- Stability margin is the maximum amount of additional gain M that can be introduced (equally and simultaneously) in each direction of the transmission without causing singing



$L_s \geq 0$

$$L_s - 2M = 0$$

$$2(B + L_2) - 2M = 0$$

$$\boxed{M = B + L_2 \quad dB}$$

الاشتقاق والقانون لـ M must be Known in the Exam

Telephone System

- A rating system was standardized by CCITT to grade a customer satisfaction is called Overall Loudness Rating (OLR) or Overall Reference Equivalent (ORE) in dB

$$ORE = TRE + RRE + losses \quad dB$$

-ve Value → Better than the Referance

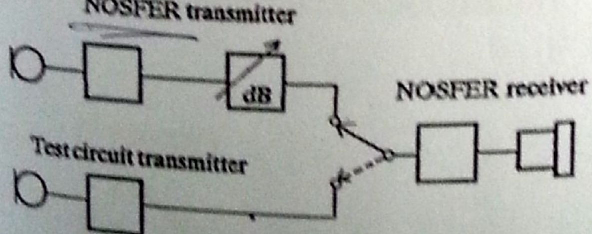
- TRE: transmit reference equivalent
- RRE: receive reference equivalent
- -ve dB: the system is better than the reference

Transmission Performance In Telephone System

- TRE and RRE is measured using a reference system called NOSFER in ITU lab in Geneva

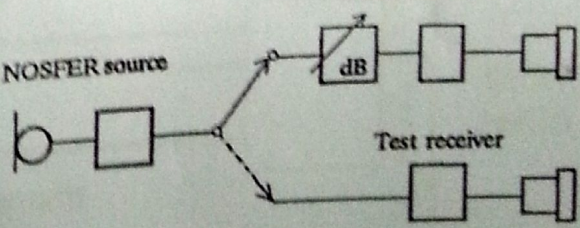
International telephony Union: to standrize the telephones manufactring. (Referance)

Ref



Transmit reference equivalent

TRE

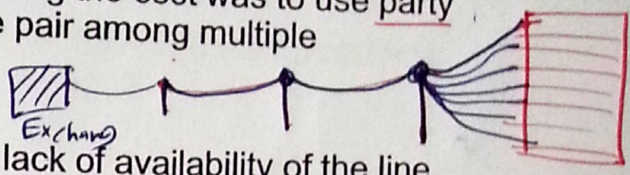


Receive reference equivalent

Pair-Gain Systems

no privacy

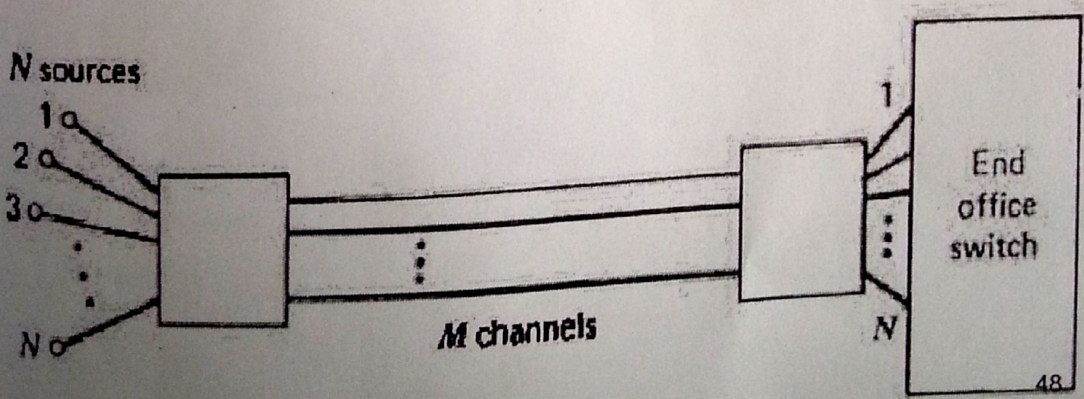
- In the past, a common means of reducing the cost was to use party lines, which involves sharing of a wire pair among multiple households.



- But due to the lack of privacy and the lack of availability of the line, the pair-gain system was used as a solution.
- **Basic types of pair-gain systems:**
  1. concentrators (remote switches)
  2. multiplexers (carrier systems)

Pair-Gain Systems

- **Concentrator**
- Since a concentration is in capable of simultaneously connecting all stations it services, a certain a mount of blocking is necessarily introduced by concentration



$M < N$

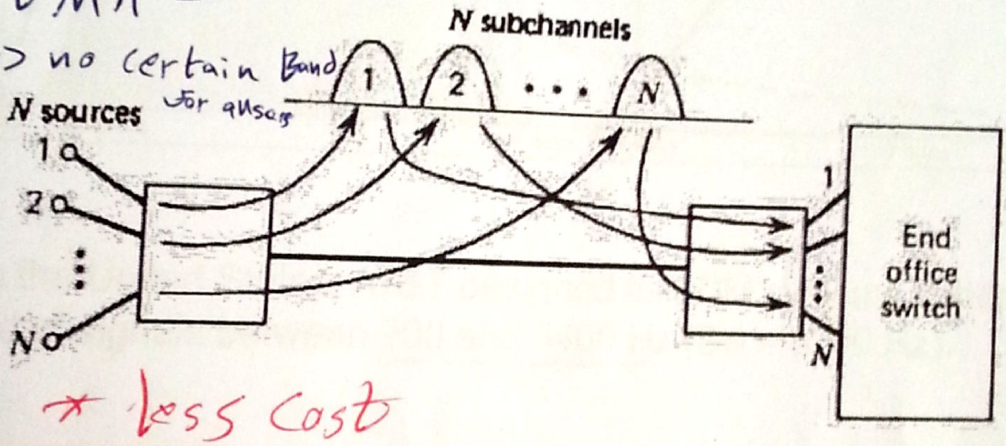
* Good in privacy
* → Blocking Probabilty → less than Party lines

Pair-Gain Systems

- **Multiplexing:**
- Space division multiplexing involves nothing more than bundling more than one pair of wires into a single cable.
- FDM → Carriers
- In digital systems (TDM)

C_1 C_2 C_3 C_4

CDMA
OFDMA → in cell
↳ no certain Band for allusers

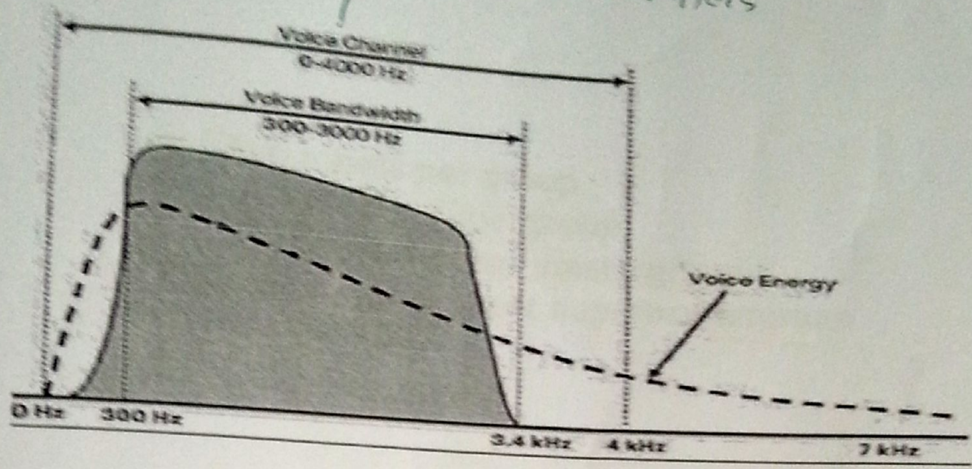


* less Cost

* less Blocking probability

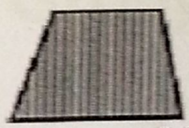
4 KHz Human Voice Channel

no sharP filters

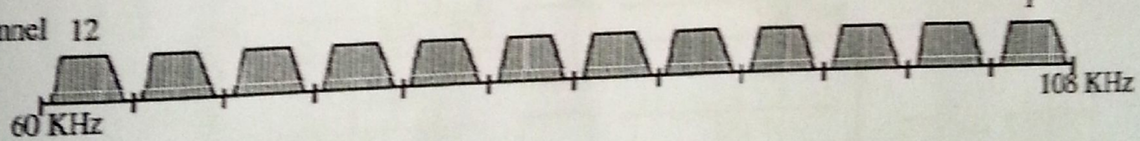


Note: In the United States, AT&T designed its FDM systems to handle the band of signals between 200 and 3400 Hz (Bw = 3200 Hz).

A Single Voice Channel [SSB]



A 12 Channel Group



- 900 Hz guardband between channels
- Single Sideband suppressed carrier modulation SSB

1) Less power in transmission

less power Utilization

2) less Bandwidth

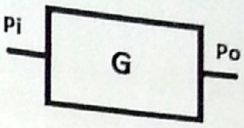
Power Levels

1dB = 10 log $\frac{x}{y}$

$\frac{1}{10}$ = log $\frac{x}{y}$

10

$$G = 10\log\frac{P_o}{P_i} \quad \text{dB}$$



$$G = 10\log\frac{P_o}{1\,\text{mW}} \quad \text{dBm}$$

used to indicate power levels relative to 1 mW

Example: 1 W = 30 dBm

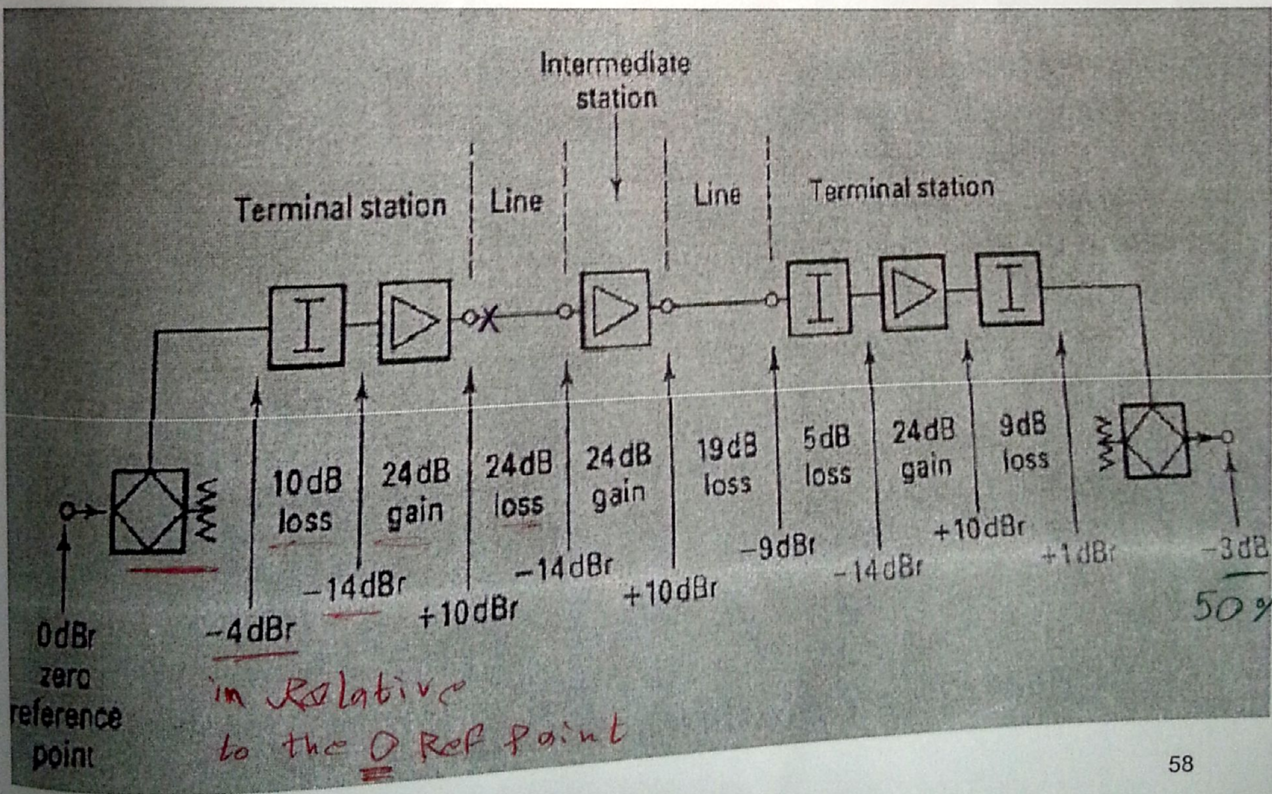
$$G = 10\log\frac{P_o}{1\,\text{W}} \quad \text{dBW}$$

used to indicate power levels relative to 1 W

P^5

30

10 log 1000



50% loss

in Relative to the 0 Ref point

dBmo → is the same

Power Levels

- Relative level of a signal at any point in the system with respect to its level at the reference point is denoted by dBr
- Signal level in terms of the corresponding level at the reference point is denoted by dBmO: dBmO= dBm-dBr

it's the same

at x $P_x = 10\,mW \rightarrow 10\,dB_m$

dBr at $x \rightarrow +10\,dB$

$dB_m O = dBm - dBr = 10 - 10 = 0$

$P_{ref} = 1\,mW$

Signaling

Types of signaling:

1. In-channel signaling or called Channel Associated Signaling (CAS): used in old telephone network.
2. Common-channel signaling (CCS).

1 channel
Data + signaling

how to signal

* every information control is signaling

→ 2 network → 1 Data
→ 1 signaling

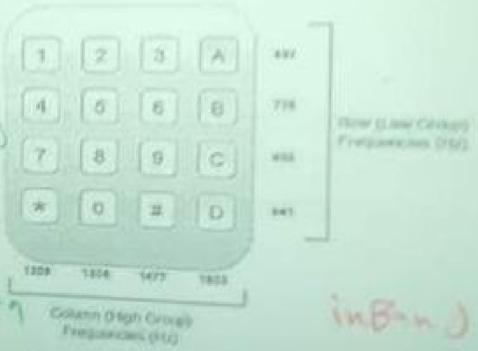
Channel Associated Signaling

a) In-band: called voice frequency signaling (VF).

- The main advantage of in-band signaling is that it can be used on any transmission medium.
- The main disadvantage is the need to eliminate mutual interference between the signaling waveforms and a user's speech.

Example: Dual-tone multi-frequency (DTMF) signals from push-button telephone.

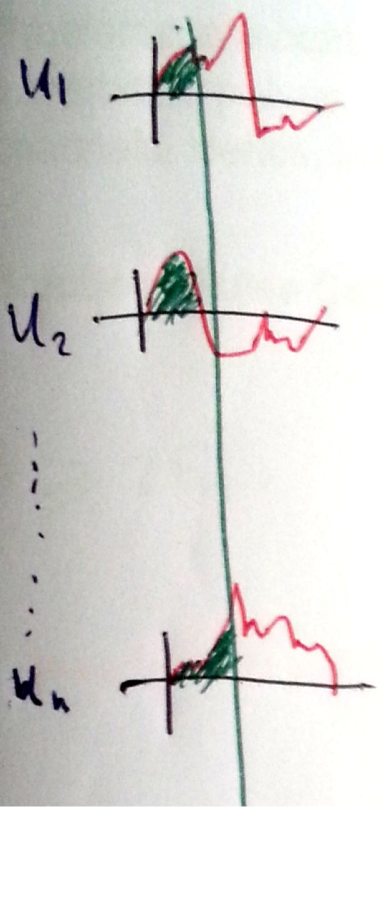
* those two signals are Unique ⇒ when Added together you can tell the Difference Between it and the Voice/noise signal



inBand

Channel Associ

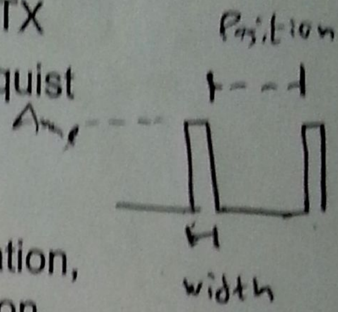
Digital Telephone Network



Time Division Multiplexing

In RX: sampling pulses should be synchronized with that in TX

In Telephone system: sampling rate is 8KHz (2 x 4 KHz, Nyquist frequency). So 8k pulses/second



Possible modulation methods: PAM-pulse amplitude modulation, PWM-pulse width modulation , PPM-pulse position modulation

Problems: attenuation and delay causes dispersion of the transmitted pulses. So pulses interfere with other pulses of adjacent channels. Hence, inter-channel crosstalk

Sample levels

Solution: Pulse Code Modulation

→ code $2^n = L = \#$ quantization levels

67

$F_s \geq 2F_m$

Frame duration $= \frac{1}{8k}$

→ easy to recover the signal

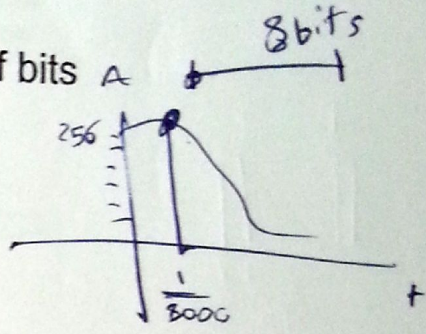
1 sample from each user / sample

* threshold

PCM

PCM

- A/D: an analog level of voltage is converted to a group of bits (word=32 bits or byte= 8 bits)



- In telephone system: 8 bit encoding is used (256 levels).
- 8 K pulse /sec (1 pulse = 8 bits)
- Bit rate = 8 k x 8 /sec = 64 kbps (= baud rate, since bit is one symbol).
- Nyquist showed that the minimum bandwidth needed to transmit a digital signal at B bauds is B/2.
- So, in telephone system: minimum BW= 32 KHz. (but for analog BW =4 KHz)

Q
Exam

PCM

- PCM introduces quantization distortion which is not found in analog transmission.
- If quantization is done using uniform size steps, then high quantization error. So, non-uniform size steps are needed **(problem).**
- It is better to assign small quantization interval for small signals and [illegible] large signals.

[illegible] and then expanding)

Examples : if a signal has an Absolute level of
1
-6 dBm at a point where the Relative level is -10 dBr , find the signal level Referred back to the zero reference point (dBm0)

Sol :

$$dBm0 = dBm - dBr$$

$$-6 - (-10) = 4\ dBm$$

Exmple
2

$P_{in} = 16.7MW \longrightarrow \boxed{AmP} \longrightarrow P_{out}\ 30mW$

find Gain in dB

Sol :

$$G = \frac{P_{out}}{P_{in}} = \frac{30m}{16.7M} = 1.8 \times 10^3$$

$$Gain\big|_{dB} = 10 \log(1.8 \times 10^3) = 32.6\ dB$$

Example : express the Fallowing power levels in dBm and dBW

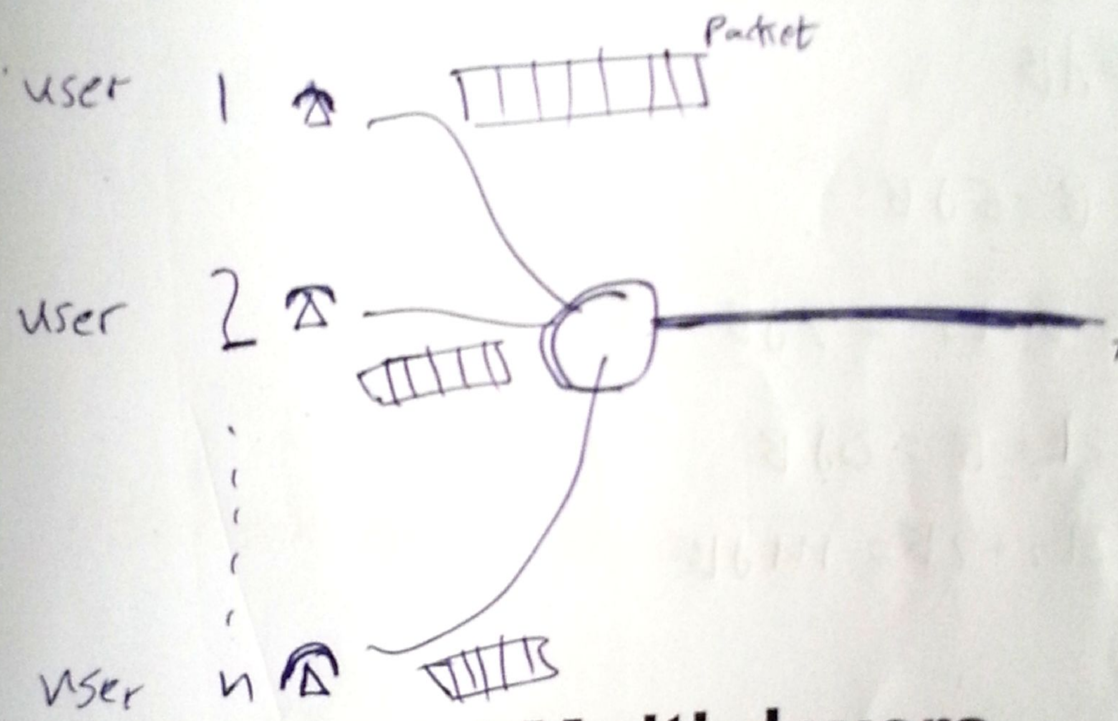
① 1mW → 0 dBm, -30 dBW

100 mW → 20 dBm, -10 dBW

② 1 W → 30 dBm, 0 dBW

TDM and VoIP

- Once a connection is established, capacity is saved even when the device is not sending information.
- But there are small slices of silence in voice (wasting network capacity).
- This is the reason TDM is being gradually replaced in high-traffic portions of networks by Voice over Internet Protocol (VoIP) technologies



Statistical Multiplexers

do not guarantee capacity for

Ex: A Four wire has an over All loss (two wire to two-wire) of 1 dB and the Balance Return loss at each End is 6 dB

1) the signaling point

2) the stability Margin

3) the attenuation of talker and listener echo

$L_2 = 1\,dB$

$B = 6\,dB$

1. $S = B = 6\,dB$
2. $m = B + L_2 = 7\,dB$
3. $L_t = 2L_2 + B = 8\,dB$

   $L_L = 2L_2 + 2B = 14\,dB$

Scanned by CamScanner

PCM Primary Multiplex Group

- European 30-channel system (E1-European system 1)
- 24- channel system (T1-*transmission system* 1) used in North America and Japan.

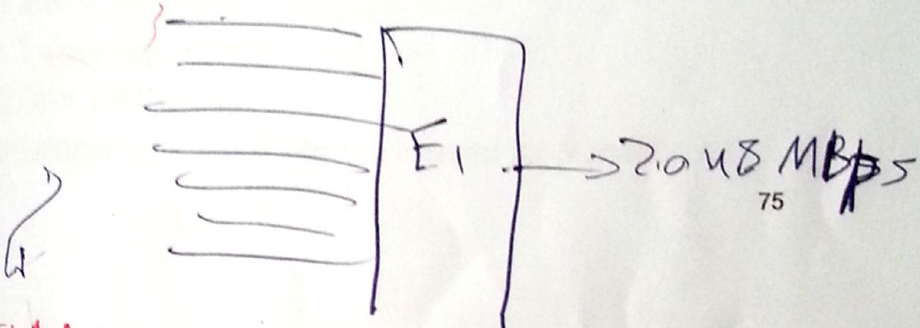
E1

- Note: T0 and E0- are used for one channel (64 Kbps)
- What about E2, E3, or T2, T3 ?

T1

30 Users ~~is using on~~ channel in the multiplixer

E1



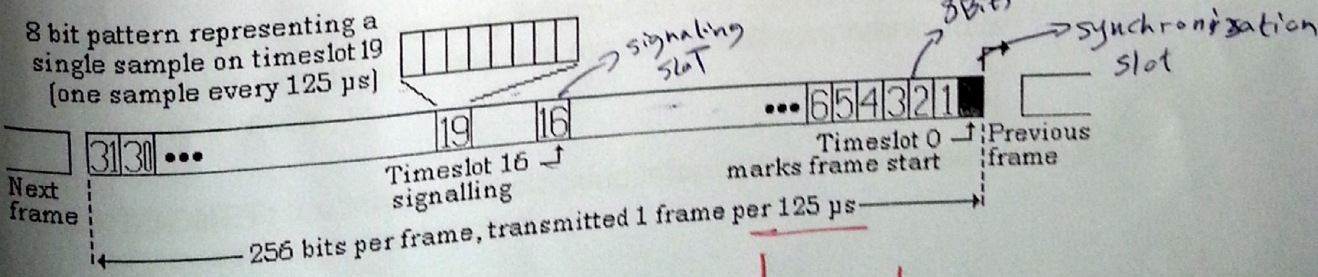
32*64 Kbps = 2.048 Mbps

E1 System

$F_s = 2F_m = 2 \times 4\,KHz$
$= 8\,K$ sample/s
Bit Rate $= 8 \times 8 = 64$ Kbps
Frame Rate $= \frac{1}{8K} = 125\,\mu s$

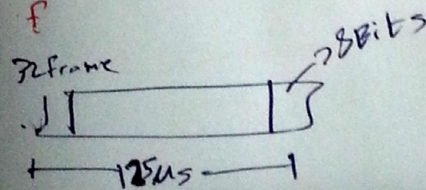
- Uses A-law companding
- Bit rate= 32*8/125=2.048 Mbps

2.048 Mbit/s digital frame format



$\frac{1}{2F_m} = \frac{1}{2 \times 8K}$

f



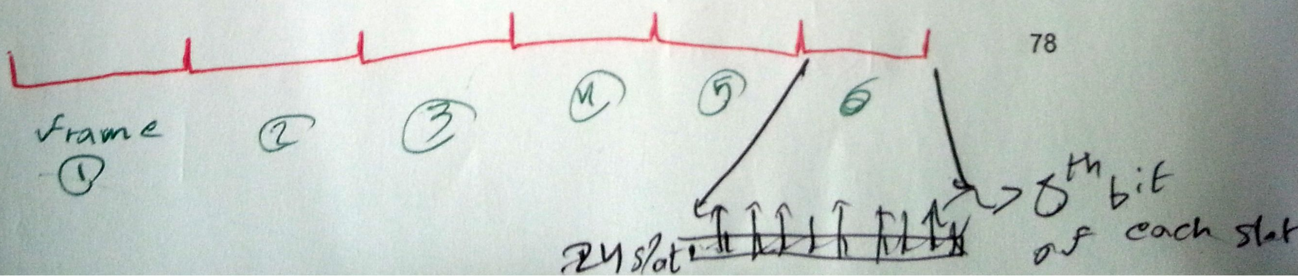
Digital Hierarchy in E1 and T1

- The plesiochronous digital hierarchy-PDH (old)
- The synchronous digital hierarchy-SDH (new)
- The synchronous optical network –SONET (new)

PDH: the timing and clocking information are contained within the digital bit stream and thus this system is self-synchronized or asynchronous.

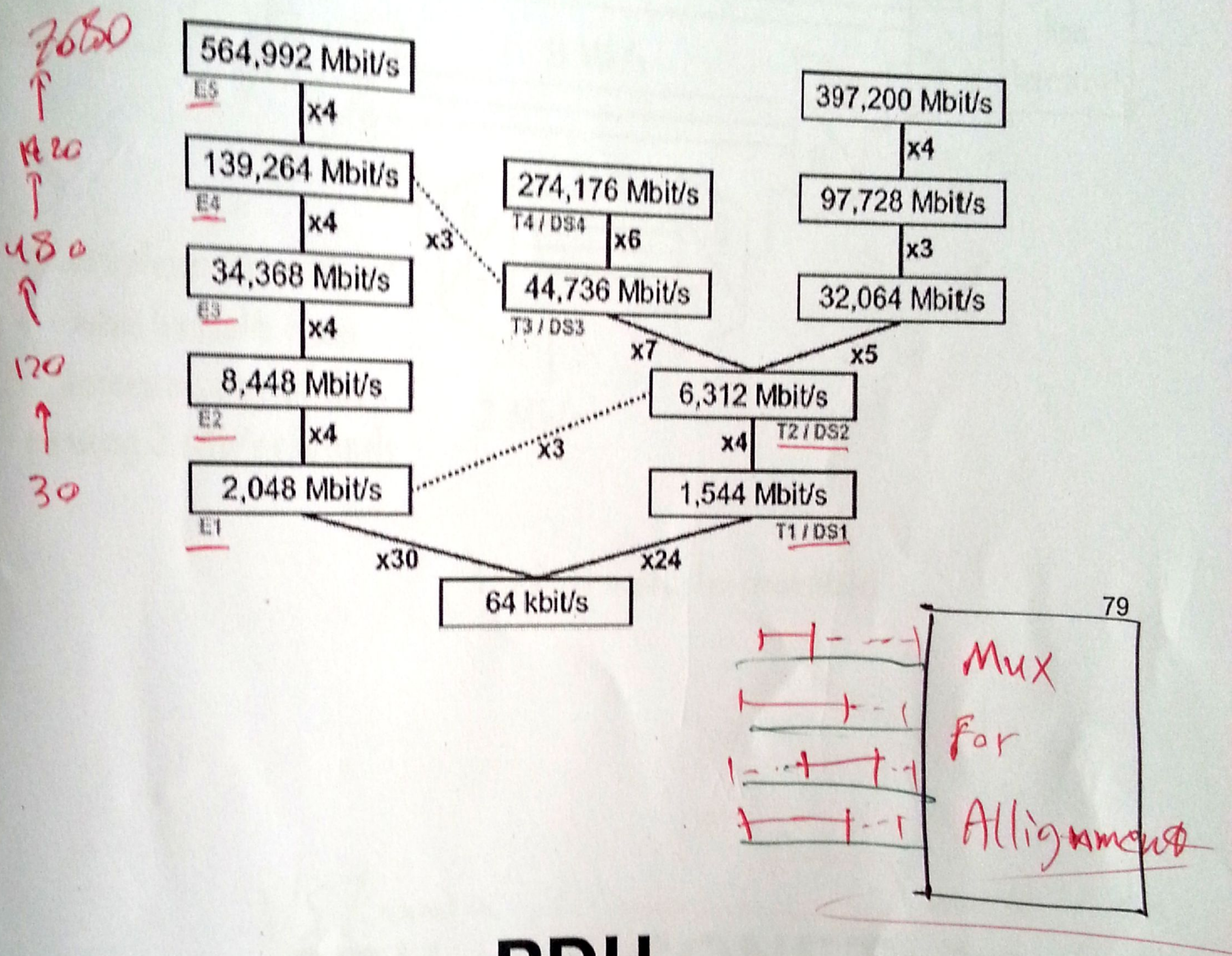
larger networks

SDH/SONET: the timing and clocking information are obtained from a highly accurate master clock



PDH

- European PDH (Ex)
- North American PDH (Tx or DSx)



PDH

1st Protocol

US/Canada   world wide

SDH and SONET → protocols

more popular

- They are standardized protocols that transfer multiple digital bit streams over optical fiber.
- Developed to replace the PDH system for transporting large amounts of telephone calls and data traffic over the same fiber without synchronization problems (synchronization sources of various circuits were different).
- SONET in the United States and Canada, and SDH in the rest of the world. Although the SONET standards were developed before SDH, it is considered a variation of SDH because of SDH's greater worldwide market penetration.

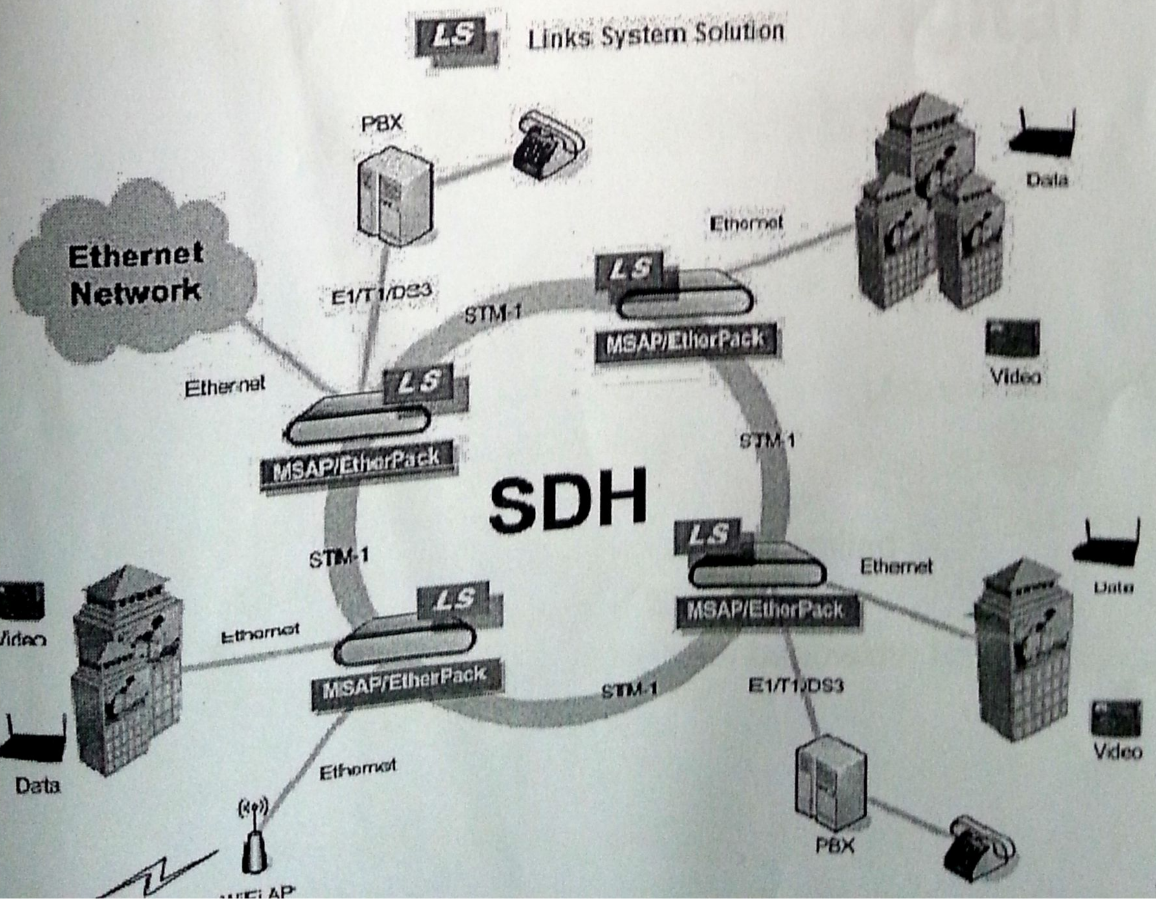
SDH and SONET

STM-1: Synchronous Transport Module, level 1
STS-1: Synchronous Transport Signal, level 1

SONET and SDH transmission speeds

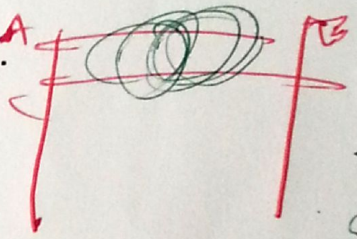
SONET signal	SDH signal	Bit rate [Mbps]
STS-1		51.84
STS-3	STM-1	155.52
STS-12	STM-4	622.08
STS-24		1244.16
STS-48	STM-16	2488.32
STS-192	STM-64	9953.28
STS-768	STM-256	39814.32

Sonet is a virsion of SDH



Advantages of Digital Voice Network

1. Ease of multiplexing → TDM Devices Cheaper than PDM Devices
2. Ease of signaling
3. Use of modern technology → Logic Gates
4. Integration of transmission and switching
   Analoge Voice channel interfer with signaling
5. Signal regeneration Repeaters
6. Performance monitorability
   no special Devices
7. Accommodation of other services VOIP + net
   → in Analoge SNR is higher
8. Operability at low signal-to-noise/interference ratios → Digital + Repeaters
9. Ease of encryption Scrambler
10. Digital signals are highly resistant to crosstalk.



D cant Recover due to synch of frames

A → B

Disadvantages Of Digital Voice Networks

1. Increased bandwidth. 4 kHz . to 64 kbps
2. Need for time synchronization.
3. Topologically restricted multiplexing. → Far Users and Near Users
4. Need for conference/extension bridges. → كشان يوزع كلهم الUsers
5. Incompatibility with analog facilities. → needs A/D & D/A
6. The information capacity of digital system is limited .
   - Shannon limit for information capacity
   - C = 3.32 B log2 (1+ S/N) , C: bps, B: bandwidth Hz

Signalling System No.7

- Uses out-band signaling and it is CCS system
- Called CCITT signaling system No. 7
- Called Common Channel Signaling System 7
- Is a set of telephony signaling protocols which are used to set up most of the world's PSTN telephone calls
- The main purpose is to start and end telephone calls but then it is used for other service such as prepaid billing mechanism and SMS

Channel Assiostated signaling

CAS